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# (54) Hearing aid system

(57) A hearing aid system includes an earpiece for producing sound representative of a listening environment. A module worn by the hearing aid user is in wireless communication with the earpiece. A plurality of microphones are located in the module to receive incoming sound signals from the listening environment. Processing circuitry in the module analyzes the incoming sound signals received by the plurality of microphones and generates control signals to govern the production of sound by the earpiece.

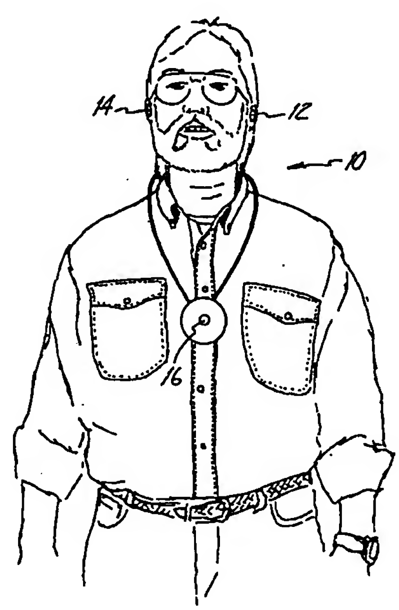


Fig. 1

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## Description

### BACKGROUND OF THE INVENTION

**[0001]** The present invention relates to a hearing aid system, and more particularly to a personal hearing aid and information system that employs beamforming and other processing techniques to optimize the performance of the hearing aid system in a variety of sound environments.

**[0002]** Hearing aids have become a popular and effective tool for enabling a person with a hearing deficiency to detect and recognize audible sounds by filtering and amplifying the sounds and reproducing them in a microphone worn in the ear of the user. Hearing aid devices have become more and more sophisticated, and binaural hearing aid systems have been developed so that a user wearing a hearing aid in each ear may identify the direction from which a sound originated based on differences in the time of arrival of a sound at each ear and differences in loudness level between the ears.

**[0003]** Despite the improvements in hearing aid technology and hearing aid system processing, existing hearing aid systems are still unable to approximate the selective listening patterns that are generated by the human brain in response to particular sound environments. Specifically, hearing aid systems are typically unable to effectively "focus" the attention of the listener onto a particular sound source where other sounds are also present. There is a continuing need in the art for a complete hearing aid system providing improved ability to discriminate among sound levels and sources.

### BRIEF SUMMARY OF THE INVENTION

**[0004]** The present invention is a hearing aid system that includes at least one earpiece and a module in wireless communication with the earpiece. A plurality of microphones are located in the module to receive incoming sound signals from a listening environment. Processing circuitry in the module analyzes the incoming sounds signals received by the microphones and generates control signals that direct the earpiece to produce sounds representative of the listening environment. The processing and analysis amplify sound signals originating from desired directions and attenuate sound signals originating from undesired directions, resulting in superior listening performance.

### BRIEF DESCRIPTION OF THE DRAWINGS

**[0005]**

FIG. 1 is a diagram illustrating a human outfitted with a hearing aid system according to the present invention

FIG. 2 is a diagram illustrating the components of a hearing aid system according to a first embodiment of the present invention.

FIG. 3 is a diagram illustrating the components of a hearing aid system according to a second embodiment of the present invention.

FIG. 4 is a block diagram illustrating the functional components of a medallion utilized in the hearing aid system of the present invention.

FIG. 5 is a block diagram illustrating the functional components of an earpiece utilized in the hearing aid system of the present invention.

FIG. 6 is a diagram illustrating a simplistic example of a multi-element sensor array for receiving incoming sound signals.

FIG. 7 is a stacked graph illustrating a beamforming example where the left and right sensor signals are in phase.

FIG. 8 is a stacked graph illustrating a beamforming example where the left and right sensor signals are 180 degrees out of phase.

FIG. 9 is a stacked graph illustrating a beamforming example where the left and right sensor signals are less than 180 degrees out of phase.

FIG. 10 is a graph illustrating the response of a hearing aid system employing two omni-directional microphones 5.25 inches apart where incoming sound signals have a frequency of 1275 Hz.

FIG. 11 is a graph illustrating the response of a hearing aid system employing two omni-directional microphones 5.25 inches apart where incoming sound signals have a frequency of 1000 Hz.

FIG. 12 is a graph illustrating the response of a hearing aid system employing two omni-directional microphones 5.25 inches apart where incoming sound signals have a frequency of 1500 Hz.

FIG. 13 is a graph illustrating the response of a hearing aid system employing two directional microphones 5.25 inches apart where incoming sound signals have a frequency of 1275 Hz.

FIG. 14 is a diagram illustrating an exemplary beam pattern exhibited by a microphone array and beamformer according to an embodiment of the present invention.

FIG. 15 is a diagram illustrating the exemplary beam pattern shown in FIG. 14 in relation to a first exemplary sound environment.

FIG. 16A is a graph illustrating an uncorrupted input signal from the speaker shown in FIG. 15 to be listened to with the hearing aid system of the present invention.

FIG. 16B is a graph illustrating the input signal of FIG. 16A corrupted with random noise.

FIG. 16C is a graph illustrating the beamformer output of the hearing aid system of the present invention.

FIG. 17 is a diagram illustrating the exemplary beam pattern shown in FIG. 14 in relation to a second exemplary sound environment.

FIG. 18A is a graph illustrating an uncorrupted input signal from the speaker shown in FIG. 17 to be listened to with the hearing aid system of the present invention.

FIG. 18B is a graph illustrating the input signal of FIG. 18A corrupted with directional noise, random noise and reverberation.

FIG. 18C is a graph illustrating the beamformer output of the hearing aid system of the present invention.

DETAILED DESCRIPTION

[0006] FIG. 1 is a diagram illustrating a human equipped with a hearing aid system 10 according to the present invention. Hearing aid system 10 includes left earpiece 12, right earpiece 14 and medallion 16 worn round the user's neck or clipped onto the user's shirt, for example. As will become apparent in the detailed description of hearing aid system 10 to follow, earpieces 12 and 14 and medallion 16 are equipped with microphones, wireless transceivers, processing circuitry, or a combination of these components, enabling hearing aid system 10 to provide performance above and beyond the performance of the individual earpieces or medallion.

[0007] FIG. 2 is a diagram illustrating the configuration of a first embodiment of hearing aid system 10a. Left earpiece 12 includes FM receiver 22, and right earpiece 14 includes FM receiver 24. Medallion 16 includes microphones 26, 28 and 30 and FM transmitter 32. FM transmitter 32 is operable to wirelessly transmit signals and data to FM receiver 22 of left earpiece 12 via wireless link 24, and to FM receiver 24 of right earpiece 14 via wireless link 36.

[0008] In an exemplary mode of operation, microphones 26, 28 and 30 operate to receive incoming sound signals from the environment around the hearing aid system user. Medallion 16 includes processing means (not shown in FIG. 2) to interpret the incoming sound signals received by microphones 26, 28 and 30 and generate appropriate control signals for reproducing sounds in the ears of the user at earpieces 12 and 14. The control signals are transmitted from medallion 16 via FM transmitter 32 along wireless links 34 and 36 to FM receivers 22 and 24 in earpieces 12 and 14. Earpieces 12 and 14 are operable in response to the control signals to reproduce the appropriate sounds in the ears of the user.

[0009] FIG. 3 is a diagram illustrating the configuration of a second embodiment of hearing aid system 10b. Left earpiece 12 includes FM transceiver 22 and microphone 23. Right earpiece 14 includes FM transceiver 24 and microphone 25. Medallion 16 includes microphones 26, 28 and 30 and FM transceiver 32. FM transceivers 22, 24 and 32 are operable to wirelessly transmit signals and data between left earpiece 12, right earpiece 14 and medallion 16 on wireless links 34 and 36.

[0010] In an exemplary mode of operation, microphones 23 and 25 in earpieces 12 and 14, respectively, and microphones 26, 28 and 30 in medallion 16 operate to receive incoming sound signals from the environment around the hearing aid system user. Data representing the sound signals received by microphone 23 in left earpiece 12 and microphone 25 in right earpiece 14 is sent from FM transceiver 22 on wireless link 34 and from FM transceiver 24 on wireless link 36 to FM transceiver 32 in medallion 16. Medallion 16 includes processing means (not shown in FIG. 3) to interpret the incoming sound signals received by microphones 26, 28 and 30 and the data signal received from earpieces 12 and 14 by FM transceiver 32. The processing means generates appropriate control signals for reproducing sounds in the ears of the user at earpieces 12 and 14. The control signals are transmitted from medallion 16 via FM transceiver 32 along wireless links 34 and 36 to FM transceivers 22 and 24 in earpieces 12 and 14. Earpieces 12 and 14 are operable in response to the control signals to reproduce the appropriate sounds in the ears of the user.

[0011] FIG. 4 is a block diagram illustrating the functional components of medallion 16 utilized in hearing aid system 10 of the present invention. In an exemplary embodiment, medallion 16 houses the main processing circuitry of hearing aid system 10, in addition to an array of microphones and an FM transceiver. Microphones 26, 28, 30 and 31 are spatially arranged on medallion 16 to optimize the ability of the microphones to distinguish the direction from which an incoming sound originates. The details of this spatial arrangement may vary; a simplistic example showing the effects of different spatial arrangements is discussed below with respect to FIGS. 6-13. Any number of microphones may be provided in medallion 16, as schematically demonstrated in FIG 4. The signals received by microphones 26, 28, 30 and 31 are amplified by amplifiers 46, 48, 50 and 51, respectively. The outputs of amplifiers 46, 48, 50 and 51 are coupled to analog multiplexer and analog-to-digital (A/D) converter 52. Analog multiplexer and A/D converter 52 converts the analog signals from amplifiers 46, 48, 50, 51 into digital representations of those signals and sequences the signals into a digital data packet for input to digital signal processor (DSP) 54. In an embodiment such as hearing aid system 10b shown in FIG. 3, signals from remote microphones in earpieces worn by the user are transmitted on wireless links 34 and 36 to FM transceiver 32 in medallion 16. These signals are converted into digital form by FM radio receiver 58 and coupled as input to DSP 54.

[0012] DSP 54 implements a numerical algorithm to process the signals received from analog multiplexer and A/D converter 52 and FM radio receiver 58 in a predetermined manner. Given a desired processing scheme for the input signals, it is within the expertise of one skilled in the art to configure the DSP 54 to imple-

ment the appropriate numerical algorithm to achieve the desired processing scheme. The objectives of such processing schemes are described below with respect to FIGS. 7-13. Executing the algorithm produces control signals for generating appropriate signals in the earpieces worn by the user. The control signals are output from DSP 54 to FM radio transmitter 56 for wireless transmission via FM transceiver 32 on wireless links 34 and 36 to the respective earpieces.

**[0013]** FIG. 5 is a block diagram illustrating the functional components of earpiece 12 (or 14) utilized in hearing aid system 10 of the present invention. Incoming sound signals are received by directional microphone 23a and omni-directional microphone 23b. In an alternative embodiment, only one of directional microphone 23a and omni-directional microphone 23b is provided in earpiece 12. The signals received by microphones 23a and 23b are amplified by amplifiers 66 and 68, respectively, which are in turn coupled to analog multiplexer and analog-to-digital (A/D) converter 70. Signals from remote microphones in a medallion worn by the user are transmitted on wireless link 34 to FM transceiver 22. These signals are converted into digital form by FM radio receiver 74 and coupled as input to DSP 72.

**[0014]** As with DSP 54 shown in FIG. 4, DSP 72 implements a numerical algorithm to process the signals received from analog multiplexer and A/D converter 70 and FM radio receiver 74 in a predetermined manner. Executing the algorithm produces control signals for generating appropriate signals in the earpieces worn by the user. The control signals are output from DSP 72 directly to receiver 78 in earpiece 12, and also to FM radio transmitter 76 for wireless transmission via FM transceiver 22 on wireless link 34 to the medallion worn by the user, where the signal will be relayed to the other earpiece worn by the user.

**[0015]** In the embodiment schematically pictured in FIG. 5, DSP 72 is shown as being located in earpiece 12. In this embodiment, DSP's may be employed in the earpieces only, with a medallion worn by the user serving as an additional location for microphones in the microphone array and as an FM relay station for signals to be transmitted between earpieces. Alternatively, an additional DSP may also be located in the medallion, to distribute the processing functions between the physical components of the hearing aid system, or for redundancy purposes. In another embodiment, a DSP may only be located in the medallion (as shown in FIG. 4), with no DSP in the earpieces. Other arrangements of functional components within the physical parts of the hearing aid system are contemplated by the present invention.

**[0016]** With the components provided as shown in FIGS. 1-5, the hearing aid system of the present invention is able to achieve high performance operation and listening features that were not possible with prior art hearing aids. To understand the performance advantages enabled by the present invention, a simplified description of a multi-sensor processing technique called beamforming is beneficial. Beamforming is a technique that works by combining the signals received by an array of sensor elements, such as microphones, to generate a desired composite response. For example, a hearing aid system beamformer boosts the signal arriving from some directions and attenuates signals arriving from other directions, to enable the hearing aid system user to "focus" attention on a particular sound source. The direction of maximum response, referred to as the maximum response axis (MRA), can be steered in different directions by properly combining the signals from each sensor element.

**[0017]** FIG. 6 depicts a simple example where multiple sensor elements are positioned in a straight line. The angle perpendicular to a line that passes through the elements is defined as the "boresight" angle, or zero degrees. The physical angle of incident sound is therefore defined within a range of -180 to +180 degrees, where 0 degrees is boresight, ±180 degrees is straight behind, +90 degrees is to the right, and -90 degrees is to the left. This simple example is illustrated in FIG. 6, where microphones 80, 82 and 84 are linearly aligned at a distance d apart and an incoming sound signal having wavelength $\lambda$ creates an angle $\theta$ with the microphone line. The difference in time of the sound signals received by adjacent microphones in the array is defined by:

$$x = \frac{360{*}d{*}\sin\theta}{\lambda} \text{ degrees of phase or}$$

$$\frac{d{*}\sin\theta}{c} \text{ seconds,}$$

where c is the speed of the waves. This difference in time may be utilized to determine the direction from which sound signals are originating.

**[0018]** In one scenario, the objective is to maximize the response of the system along the boresight axis and minimize the response in all other directions. This is a simple processing scenario - the signals from the sensors need only be added together with equal weight given to the signals from each sensor. The results of this scheme are simplistically shown in FIGS. 7-9, where it is assumed that there are only two sensor elements for illustration purposes. FIG. 7 illustrates that where the incoming sound signals are in phase with each other (that is, the sound signals reach the two sensors at the same time, meaning that their origination direction is "boresight"), and the signals received by the sensors are as illustrated by curves 90 and 92, adding the signals will result in curve 94 having the sum of the amplitudes of curves 90 and 92. FIG. 8 illustrates that where the incoming sound signals are 180 degrees out of phase (that is, the sound signals reach the sensors at

times separated by a half cycle), and the signals received by the sensors are as illustrated by curves 100 and 102, adding the signals will result in a zero signal 104 - the signals completely cancel. This will occur when the sound signals are coming from an angle that is ±90 degrees from boresight. FIG. 9 illustrates that where the incoming sound signals are somewhat out of phase (that is, the sound signals reach the sensors at times separated by less than a half cycle), and the signals received by the sensors are as illustrated by curves 110 and 112, adding the signals will result in curve 114 having an amplitude that is less than the sum of the amplitudes of curves 110 and 112. The angle of arrival of the incoming sound signals can be calculated based on the difference in time of arrival, where the spacing between microphones is known.

**[0019]** It can be appreciated from the discussion above that the spacing of sensor elements is important to the ability to determine the origination direction of incoming sound signals. The ideal spacing of sensor elements is a half wavelength, so that an incoming sound signal coming from 90 degrees to boresight reaches adjacent sensors at times separated by a half cycle and the signals completely cancel one another. With this spacing, the range of combined signal amplitudes extends from 0 (complete cancellation) to twice the amplitude of the incoming signal, for the two-sensor example, which is the maximum range. Where microphones are spaced more than a half wavelength apart (typically for higher frequencies which have lower wavelengths), spatial aliasing occurs and grating lobes form in the response pattern, which makes accurate determination of the origination of the sound signal more difficult.

**[0020]** FIGS. 10-13 show the response of the two-microphone array over the full range of sound signal origination angles, with 0 degrees being boresight, -90 degrees being to the left, and +90 degrees being to the right. FIG. 10 illustrates the scenario where the two microphones are omni-directional microphones spaced 5.25 inches apart and the frequency of incoming sound signals is 1275 Hz (at 1275 Hz, 5.25 inches is a half wavelength). Curve 120 shows that the peak response of the system (adding the signals received at the two microphones) is at angles of 0 and ±180 degrees, with a deep "null" (attenuated response) at an angle of ±90 degrees. FIG. 11 illustrates the same microphone scenario where the frequency of the incoming sound signals is 1000 Hz - at this frequency, using the same adding formula, curve 130 shows that the nulls at ±90 degrees are much less definitive. FIG. 12 illustrates the same microphone scenario where the frequency of the incoming sound signals is 1500 Hz - at this frequency, using the same adding formula, curve 140 shows that side lobes are formed at ±90 degrees, with lower peak amplitudes, and the nulls are shifted to about ±60 degrees and ±120 degrees. These curves illustrate the importance of having microphone elements that are a half wavelength apart; where the microphone spacing is more or less than a half wavelength, the response curves are not as well defined, which tends to result in less ability for the user of a hearing aid system to focus on particular sounds.

**[0021]** FIG. 13 illustrates an alternative scenario where the microphones utilized in the two-element array are directional microphones, spaced 5.25 inches apart, where the frequency of incoming sound signals is 1275 Hz. Curve 150 shows that the peak response of the system is again at the boresight angle, with a lesser peak at ±180 degrees, and wide nulls in the vicinity of ±90 degrees. The use of directional microphones can potentially allow the hearing aid user to more effectively focus on frontal sounds while attenuating sounds originating from the rear.

**[0022]** FIG. 14 is a diagram illustrating an exemplary beam pattern exhibited by microphone array 200 and an associated beamformer according to an embodiment of the present invention. In the embodiment depicted in FIG. 14, microphone array 200 exhibits a large frontal or main lobe 202, representing sounds coming from the front of a user. Microphone array 200 also exhibits two side lobes 204 and 206 and a back lobe 208. The beam former outputs are such that sounds in main lobe 202 are amplified the most, while sounds in side lobes 204 and 206 are attenuated somewhat and sounds in back lobe 208 are attenuated even more. This beam pattern is designed to allow the user to effectively focus on sounds originating in front of the user (where the user is typically looking), even in an environment where random noise exists to the user's sides and rear. It should be understood that other beam patterns may be formed according to the present invention, and the pattern shown in FIG. 14 is an example of one particular beam pattern that may be implemented when desirable.

**[0023]** FIG. 15 is a diagram illustrating the exemplary beam pattern shown in FIG. 14 in relation to a first exemplary sound environment. The sound environment includes multiple sources of random noise 210 (such as a crowd of people) at several points around the hearing aid user (wearing microphone array 200), and a keynote speaker 212 in front of the user. As can be seen in FIG. 15, keynote speaker 212 is located at the center of main lobe 202, and sounds originating from speaker 212 will therefore be maximally amplified. Random noise 210 originates from locations at the sides of main lobe 202, and at side lobes 204 and 206 and back lobe 208, and therefore will be attenuated to a level below that of keynote speaker 212.

**[0024]** FIGS. 16A, 16B and 16C are graphs depicting an uncorrupted sound signal from keynote speaker 212 (FIG. 15), the sound signals received at the left and right input channels of the hearing aid system worn by the user, and the beamformer output from the hearing aid system to the ears of the user. The uncorrupted signal is shown in FIG. 16A. The sound signals received at

the left and right input channels are shown in FIG. 16B. The original input signal (shown in FIG. 16A) is received at each input channel, with a time-shift that is based on the physical distance between the left and right input channel microphones. The left and right channel signals are also corrupted by some level of random noise that persists throughout the listening time frame. The beamformer output is shown in FIG. 16C. The beamformer recognizes the characteristics of the random noise and of the original input signal, and also recognizes, based on the time-shift of the input signal at the left and right channels, which direction the input signal originates from. Since the signal originates from the user's front (see FIG. 15), the signal is amplified to a higher level, while the random noise is attenuated to a lower level, thereby permitting the hearing aid user to better distinguish between the desired sound (keynote speaker 212, FIG. 15) and random noise (noise sources 210, FIG. 15).

**[0025]** FIG. 17 is a diagram illustrating the exemplary beam pattern shown in FIG. 14 in relation to a second exemplary sound environment. The sound environment includes multiple sources of random noise 210 (such as a crowd of people) at several points around the hearing aid user (wearing microphone array 200), a keynote speaker 212 in front of the user, and a relatively loud directional noise source 214. The sound environment may also include reverberation 216 of the desired noise signal from the walls of the room. As can be seen in FIG. 15, keynote speaker 212 is located at the center of main lobe 202, and sounds originating from speaker 212 will therefore be maximally amplified. Random noise 210, directional noise 214 and reverberation 216 originate from locations at the sides of main lobe 202, and at side lobes 204 and 206 and back lobe 208, and therefore will be attenuated to a level below that of keynote speaker 212.

**[0026]** FIGS. 18A, 18B and 18C are graphs depicting an uncorrupted sound signal from keynote speaker 212 (FIG. 17), the sound signals received at the left and right input channels of the hearing aid system worn by the user, and the beamformer output from the hearing aid system to the ears of the user. The uncorrupted signal is shown in FIG. 18A. The sound signals received at the left and right input channels are shown in FIG. 18B. The original input signal (shown in FIG. 18A) is received at each input channel, with a time-shift that is based on the physical distance between the left and right input channel microphones. The left and right channel signals are also corrupted by some level of random noise that persists throughout the listening time frame, a higher level of directional noise, and reverberation of the input signal from the walls of the room. The beamformer output is shown in FIG. 18C. The beamformer recognizes the characteristics of the random noise, directional noise, reverberation, and the original input signal, and also recognizes, based on the time-shift of the input signal at the left and right channels, which direction the input signal originates from. Since the signal originates from the user's front (see FIG. 17), the signal is amplified to a higher level, while the random noise and the directional noise is attenuated to a lower level, thereby permitting the hearing aid user to better distinguish between the desired sound (keynote speaker 212, FIG. 17) and noise (noise sources 210, directional noise source 214, FIG. 17).

**[0027]** The hearing aid system of the present invention, as shown in FIGS. 1-5, provides an extensive sensor array and processing capability that enables significant flexibility and control of the response of the system to particular sound patterns. The shape of the response curves (shown simplistically in FIGS. 10-13 for a two-microphone example where the signals are merely added together) can be adjusted by performing different numerical algorithms on the signals received by the sensors. This adjustment can involve a technique called "adaptive null steering" which operates to move the nulls in the response pattern to a particular angular location, so that a particular noise source may be somewhat blocked out of the user's hearing, enabling better focus on the desired sound source. The provision of wireless communication links between earpieces and a medallion worn around a user's neck allows a number of spacings between microphones in the array to be utilized to provide the optimum half wavelength spacing to accommodate a higher range of frequencies of incoming sound signals. The hearing aid system therefore is able to provide superior performance compared to prior hearing aid devices, effectively enabling a listener to focus attention on a desired sound source in a variety of listening environments.

**[0028]** Although the present invention has been described with reference to preferred embodiments, workers skilled in the art will recognize that changes may be made in form and detail without departing from the spirit and scope of the invention.

## Claims

1. A hearing aid system comprising:

   an earpiece for producing sound representative of a listening environment;
   a module in wireless communication with the earpiece;
   a plurality of microphones located in the module to receive incoming sound signals from the listening environment;
   processing circuitry in the module for analyzing the incoming sound signals received by the plurality of microphones and generating control signals to govern the production of sound by the earpiece.

2. The hearing aid system of claim 1, further comprising:

at least one microphone located in the earpiece to receive incoming sound signals from the listening environment.

3. The hearing aid system of claim 2, wherein the at least one microphone located in the earpiece comprises an omni-directional microphone.

4. The hearing aid system of claim 2, wherein the at least one microphone located in the earpiece comprises a directional microphone.

5. The hearing aid system of claim 2, wherein the at least one microphone located in the earpiece comprises an omni-directional microphone and a directional microphone.

6. The hearing aid system of claim 2, wherein the earpiece and the module each include a wireless transceiver for effecting communication between the earpiece and the module.

7. The hearing aid system of claim 1, wherein the earpiece includes a wireless receiver and the module includes a wireless transmitter for transmitting information from the module to the earpiece.

8. The hearing aid system of claim 1, further comprising a second earpiece for producing sound representative of the listening environment, the module being in wireless communication wit the second earpiece.

9. The hearing aid system of claim 8, further comprising:

   at least one microphone located in the first-named and second earpieces to receive incoming sound signals from the listening environment.

10. The hearing aid system of claim 9, wherein the first-named and second earpieces and the module each include a wireless transceiver for effecting communication between the earpieces and the module.

11. The hearing aid system of claim 9, wherein the at least one microphone in the first-named and second earpieces comprises an omni-directional microphone.

12. The hearing aid system of claim 9, wherein the at least one microphone in the first-named and second earpieces comprises a directional microphone.

13. The hearing aid system of claim 9, wherein the at least one microphone in the first-named and second earpieces comprises an omni-directional microphone and a directional microphone.

14. The hearing aid system of claim 8, wherein the first-named and second earpieces each include a wireless receiver and the module includes a wireless transmitter for transmitting information from the module to the earpieces.

15. The hearing aid system of claim 1, wherein the processing circuitry comprises a digital signal processor for performing a numerical algorithm on the incoming sound signals received by the microphones to shape a response of the hearing aid system that is represented by the sound produced by the earpiece.

16. A method of producing sounds in the ears of a user that represent a listening environment, the method comprising:

   providing a plurality of microphones in a module worn by the user to receive incoming sound signals from the listening environment;
   processing the incoming sound signals received by the microphones to amplify sound signals originating from desired directions and to attenuate sound signals originating from undesired directions;
   generating control signals to govern the production of sound in the user's ears according to the result of the processing step; and
   communicating the control signals to earpieces located in the user's ears to govern the production of sound therein.

17. The method of claim 16, further comprising:

   providing at least one microphone in the earpieces located in the user's ears to receive incoming sound signals from the listening environment.

18. The method of claim 16, wherein the processing step comprises:

   performing a numerical algorithm on the incoming sound signals received by the microphones to modify the signals according to a predetermined scheme for locating peaks and nulls in the response of the hearing aid system.

19. The method of claim 16, wherein the communicating step comprises wirelessly transmitting the control signals from the module to the earpieces.

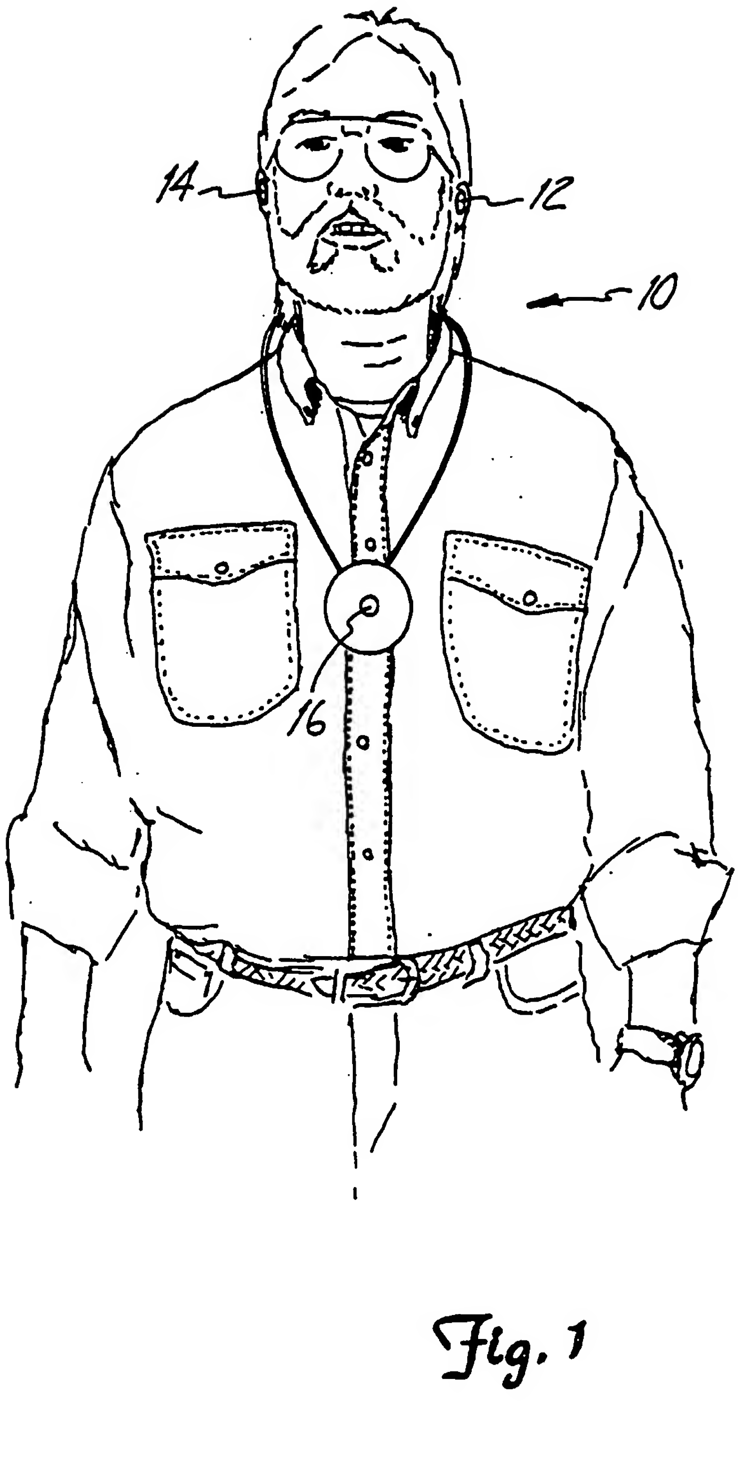



Fig. 1

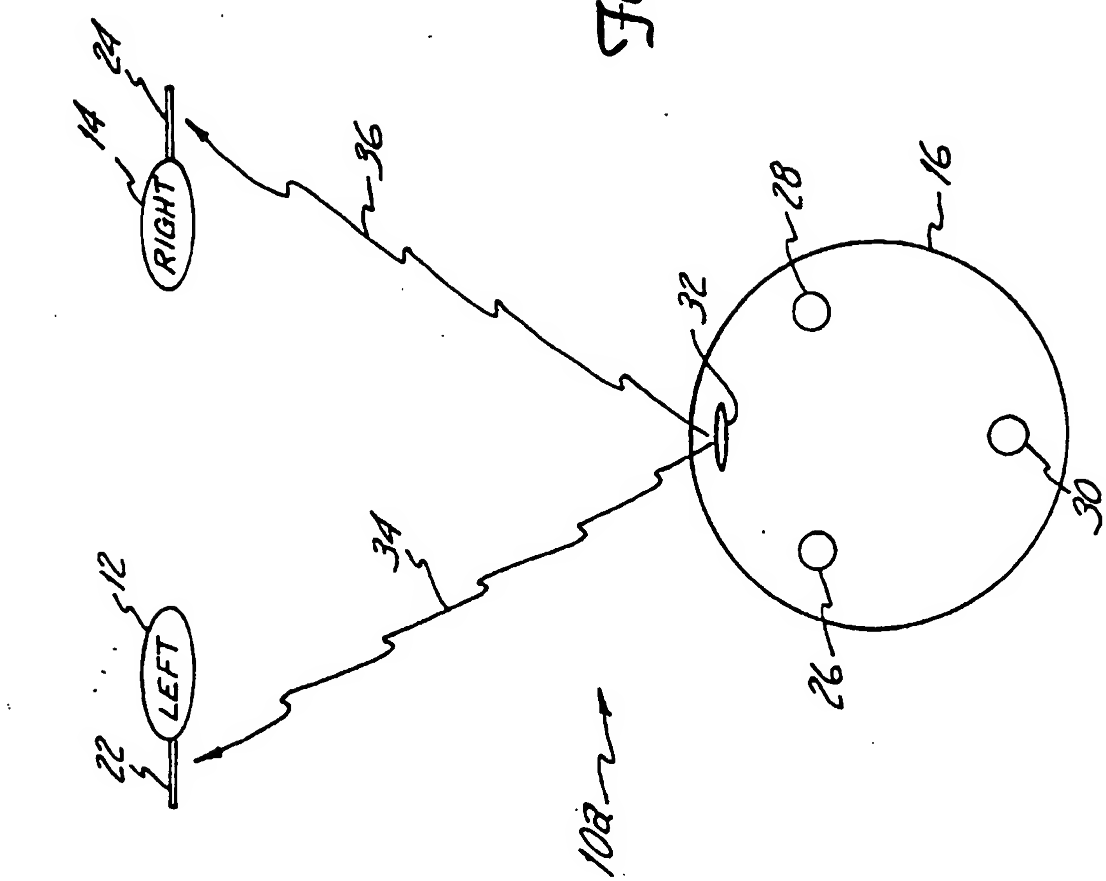



Fig. 2

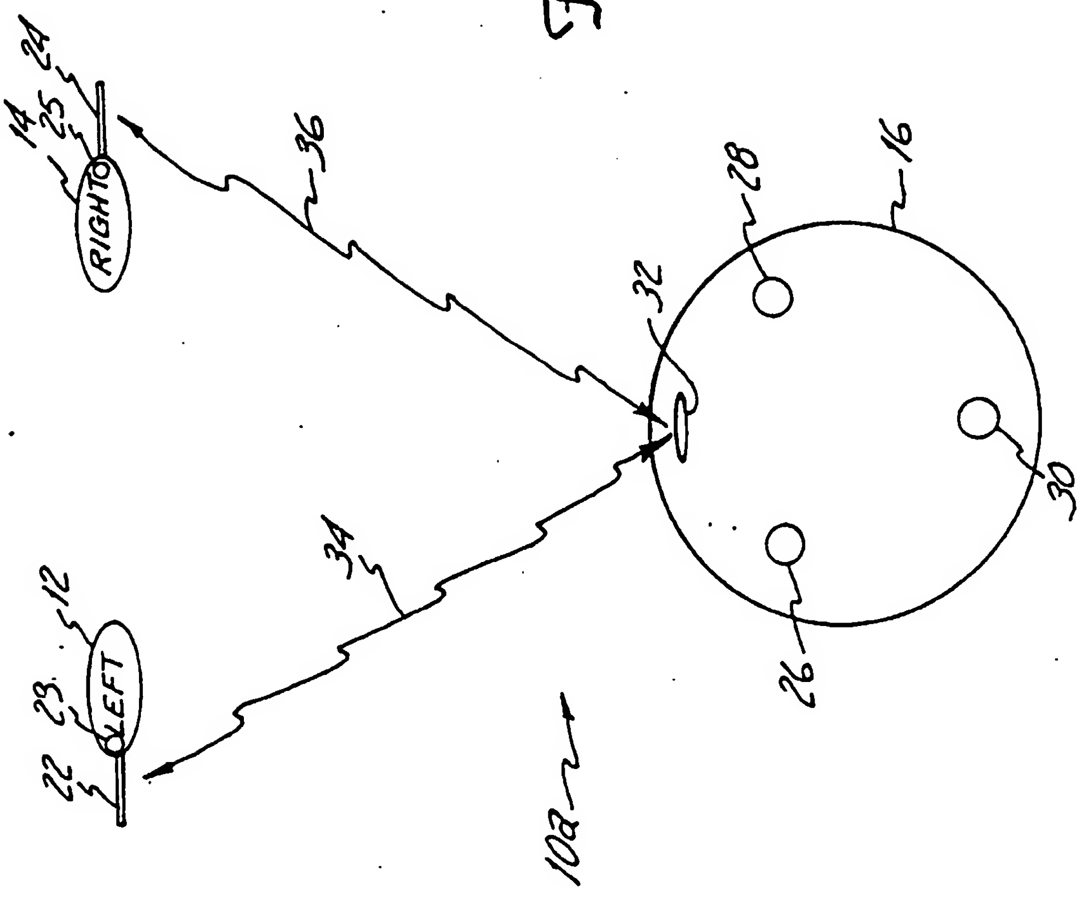
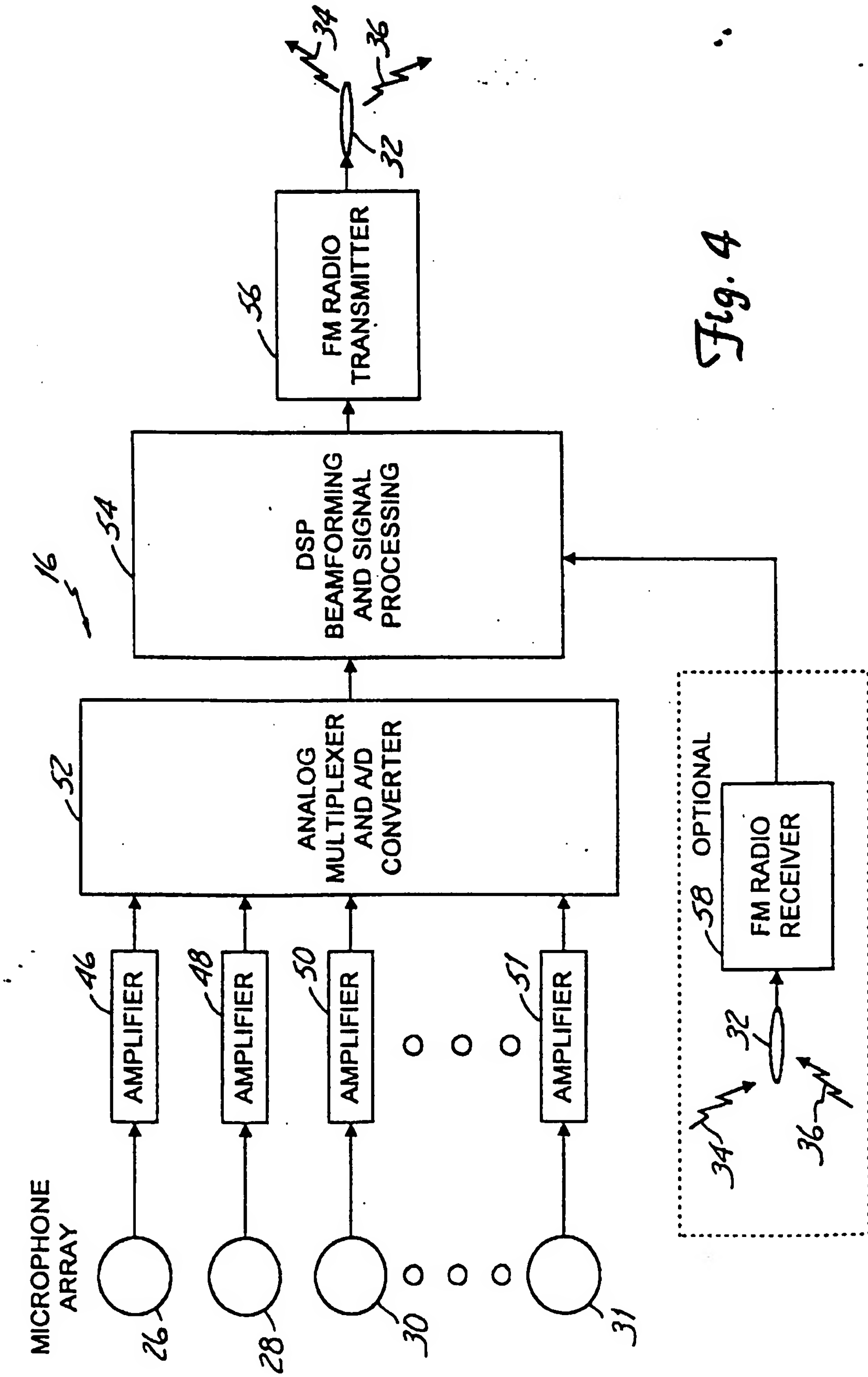



Fig.3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AMPLIFIER
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AMPLIFIER
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50
AMPLIFIER
51
ANALOG MULTIPLEXER AND A/D CONVERTER
52
DSP BEAMFORMING AND SIGNAL PROCESSING
54
16
FM RADIO TRANSMITTER
56
32
34
36
OPTIONAL
FM RADIO RECEIVER
58
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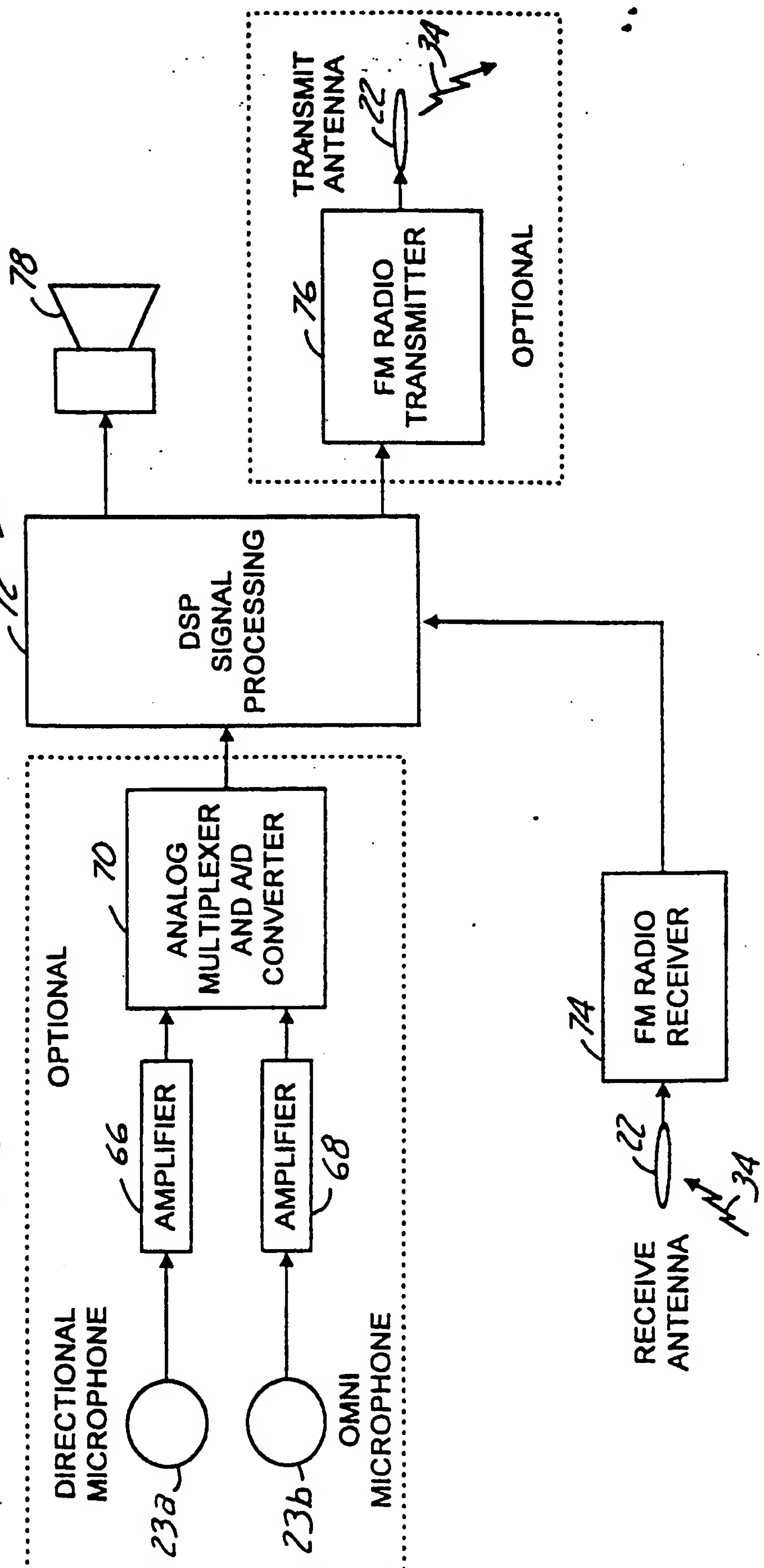
Fig. 4

Fig. 5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DIRECTIONAL MICROPHONE
23a
OMNI MICROPHONE
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AMPLIFIER
66
AMPLIFIER
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OPTIONAL
ANALOG MULTIPLEXER AND A/D CONVERTER
70
DSP SIGNAL PROCESSING
72
78
FM RADIO TRANSMITTER
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TRANSMIT ANTENNA
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34
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FM RADIO RECEIVER
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22
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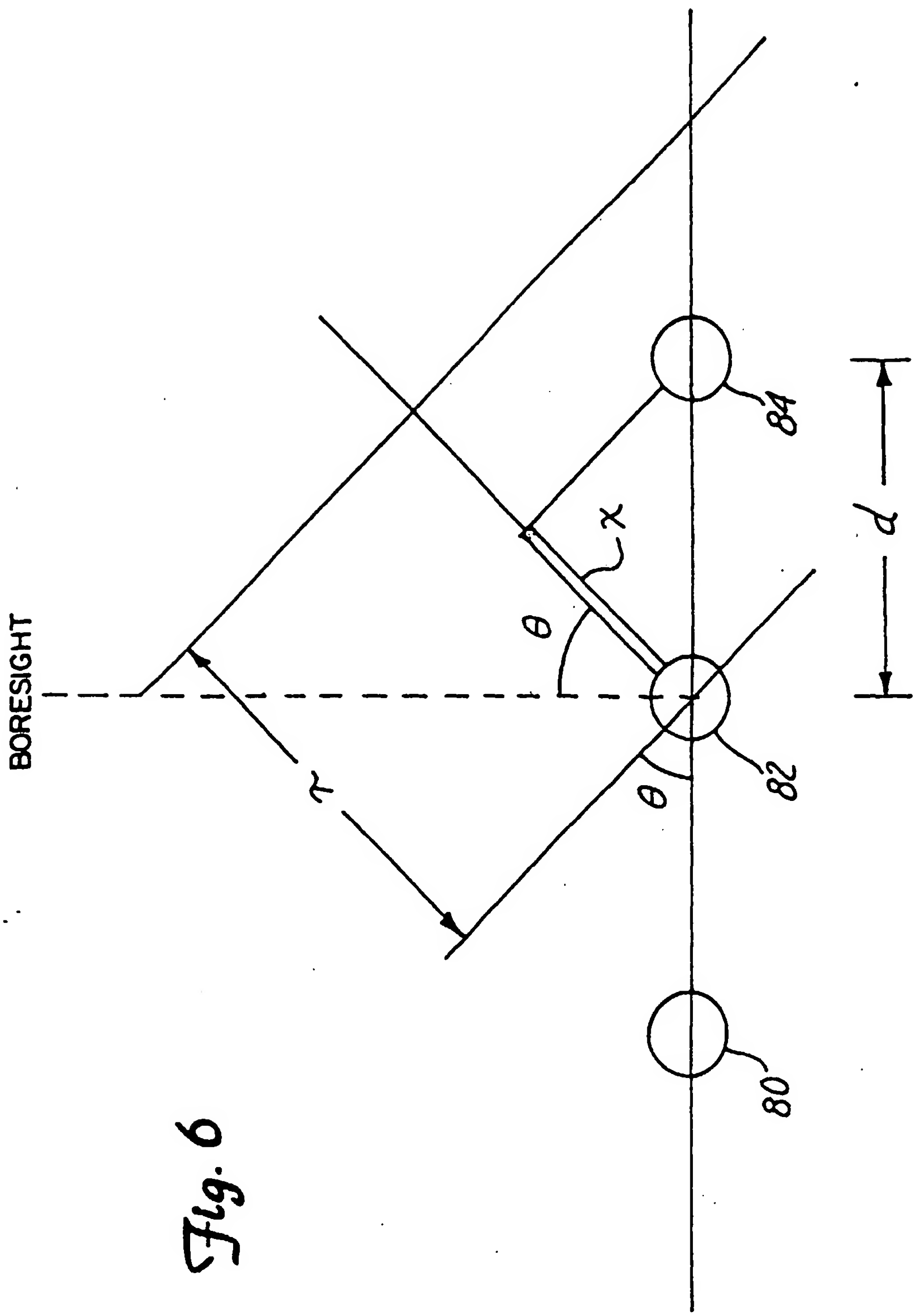



Fig. 6

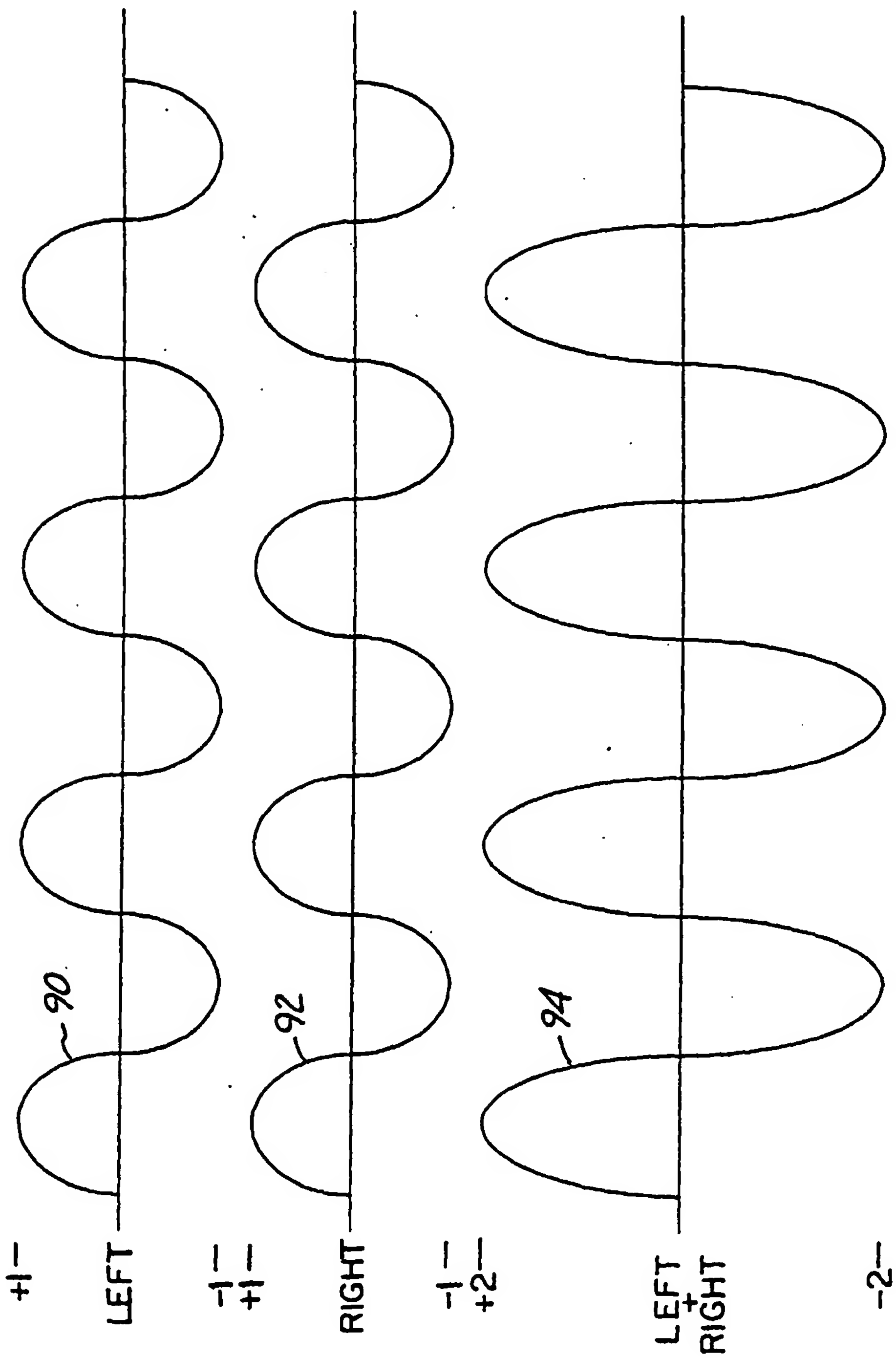



Fig. 7

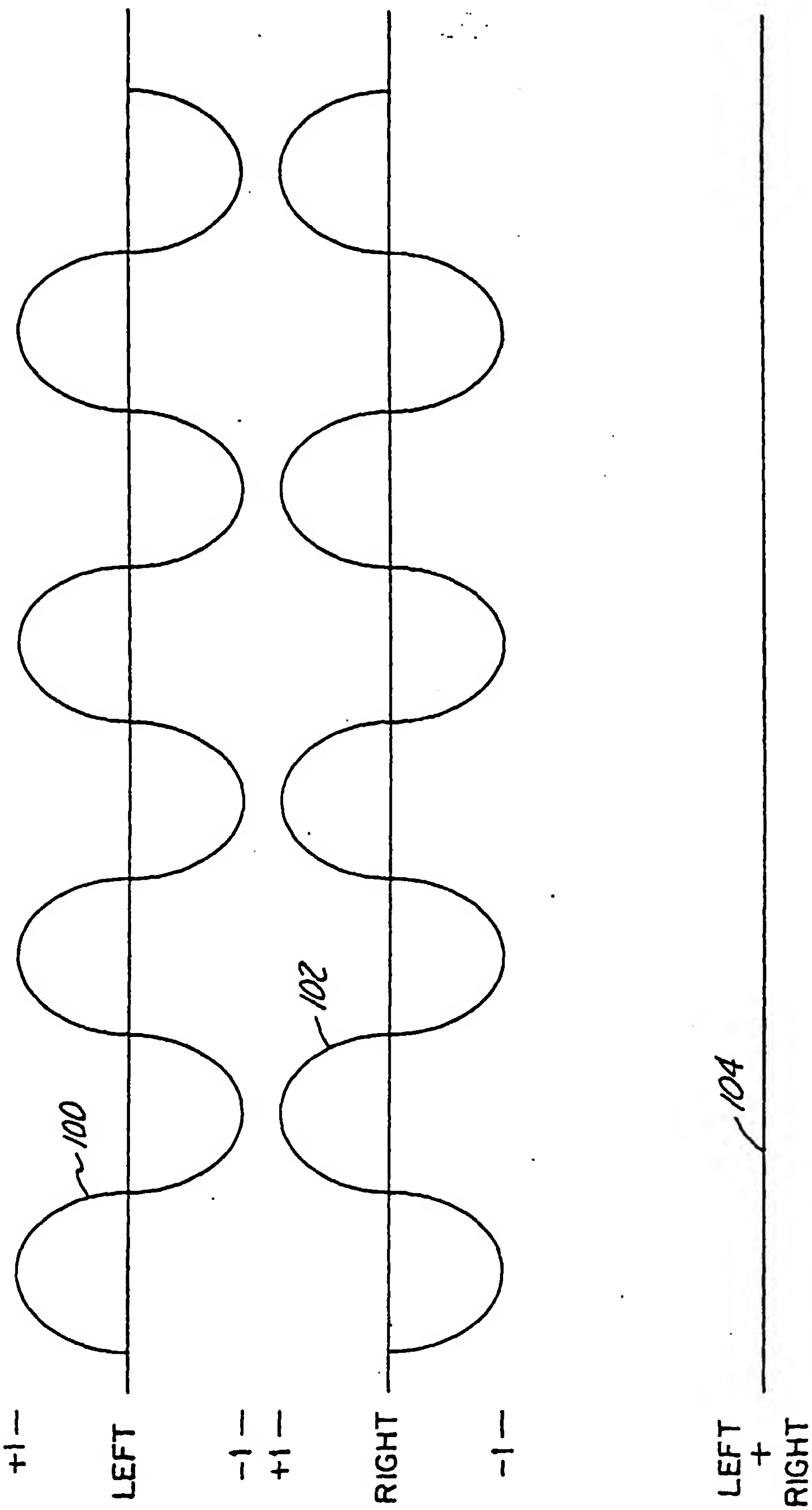



Fig. 8

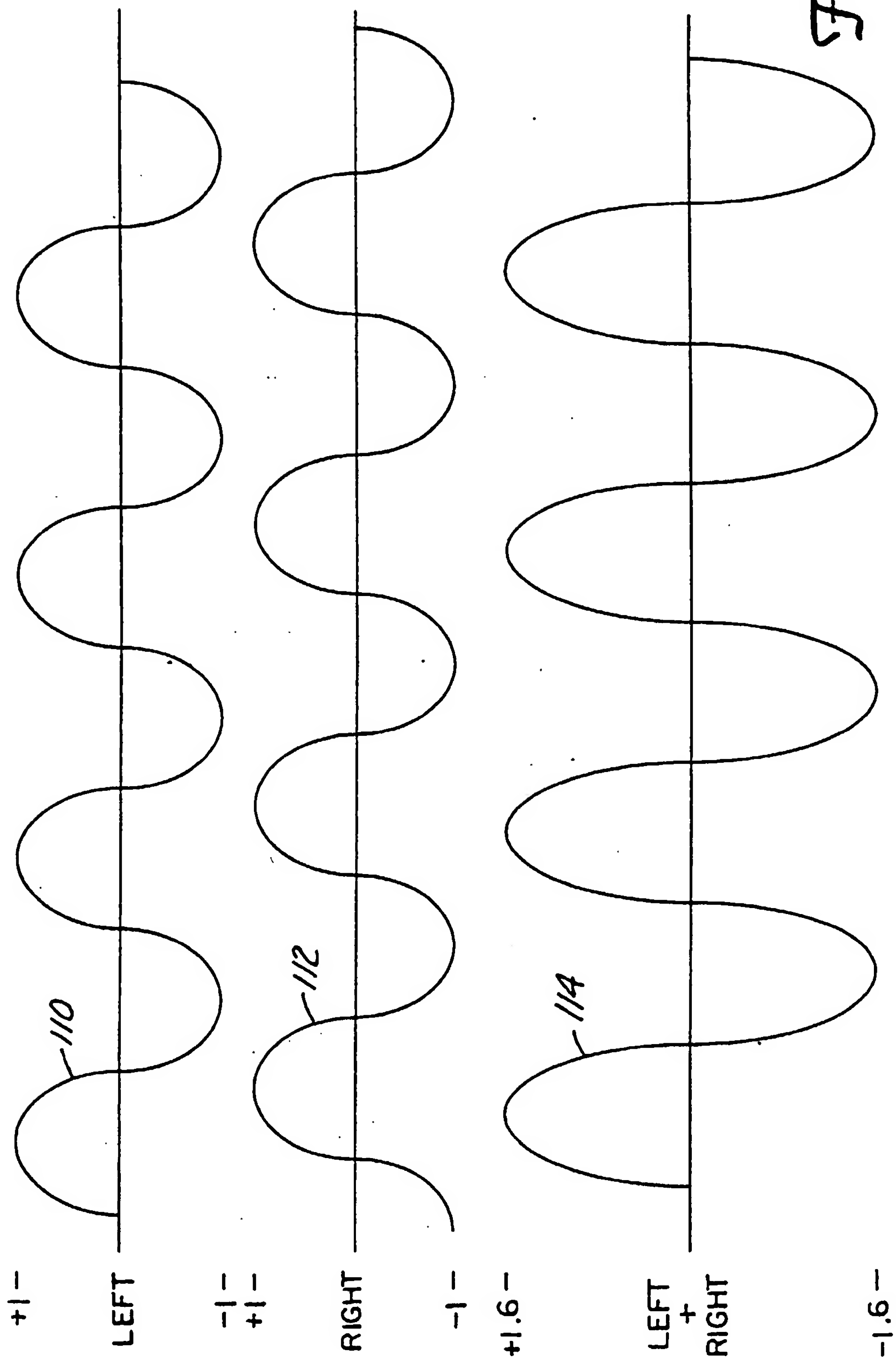



Fig.9

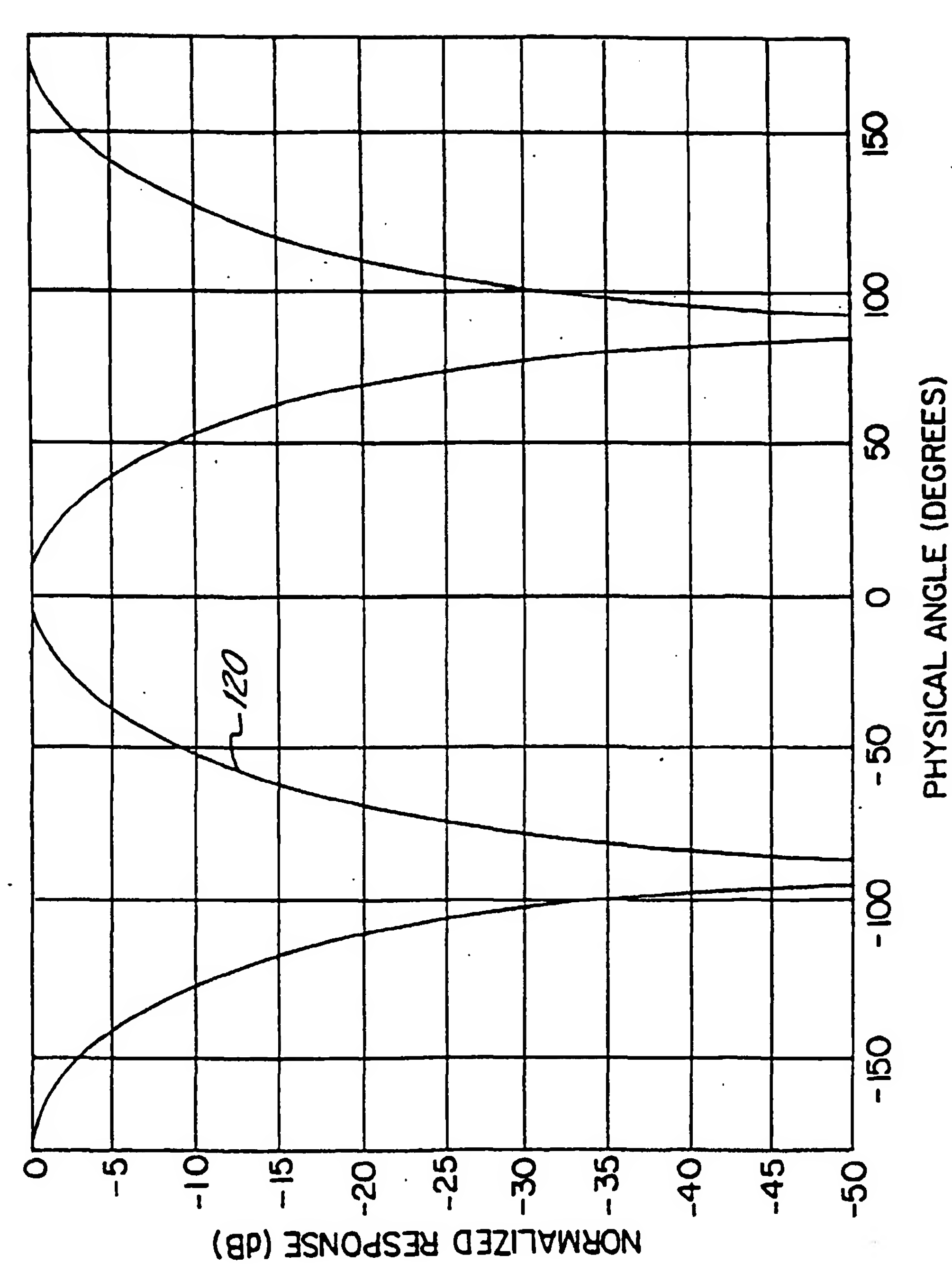

NORMALIZED RESPONSE (dB)
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
PHYSICAL ANGLE (DEGREES)
-150
-100
-50
0
50
100
150
120

Fig. 10

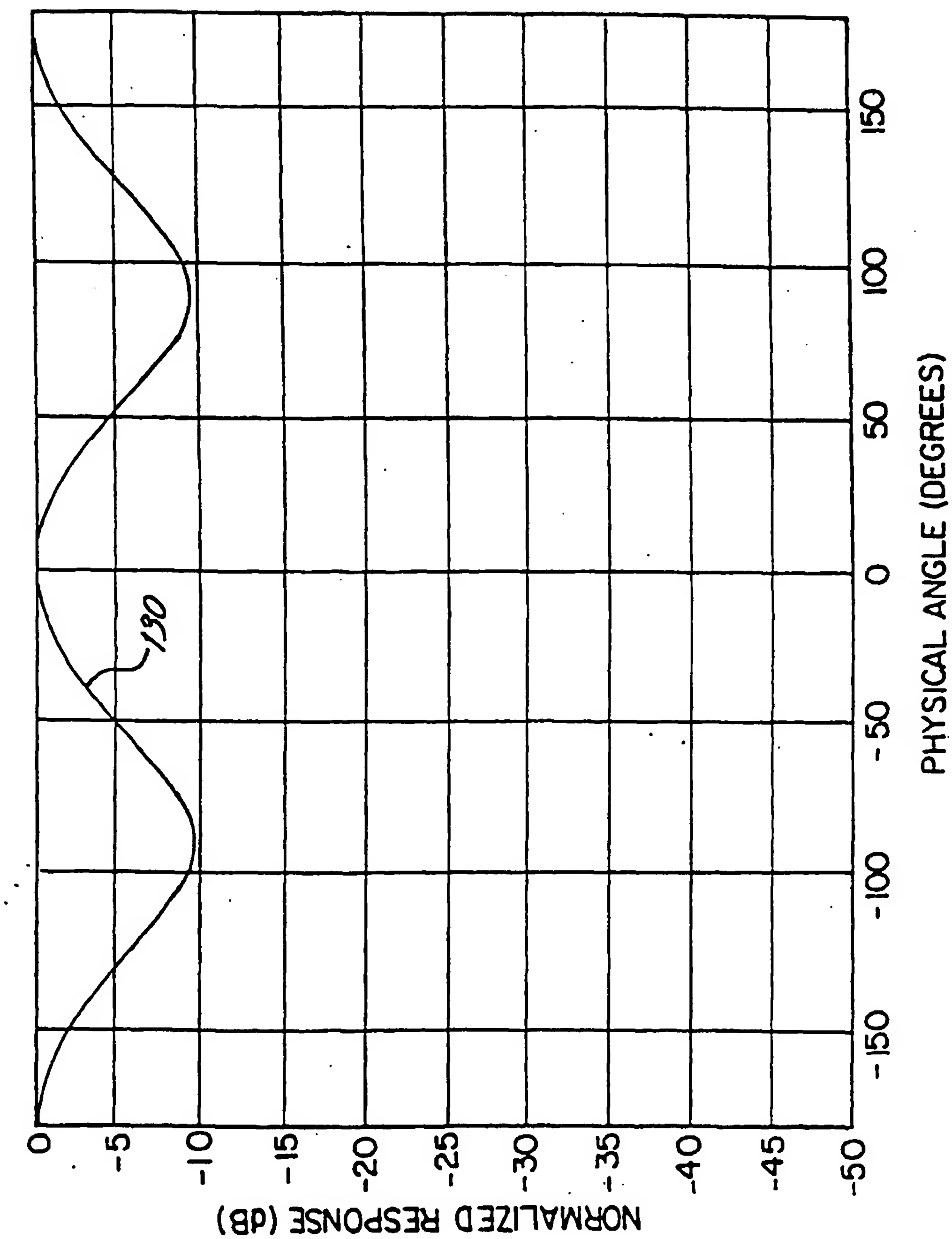

NORMALIZED RESPONSE (dB)
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
PHYSICAL ANGLE (DEGREES)
-150
-100
-50
0
50
100
150
130

Fig. 11

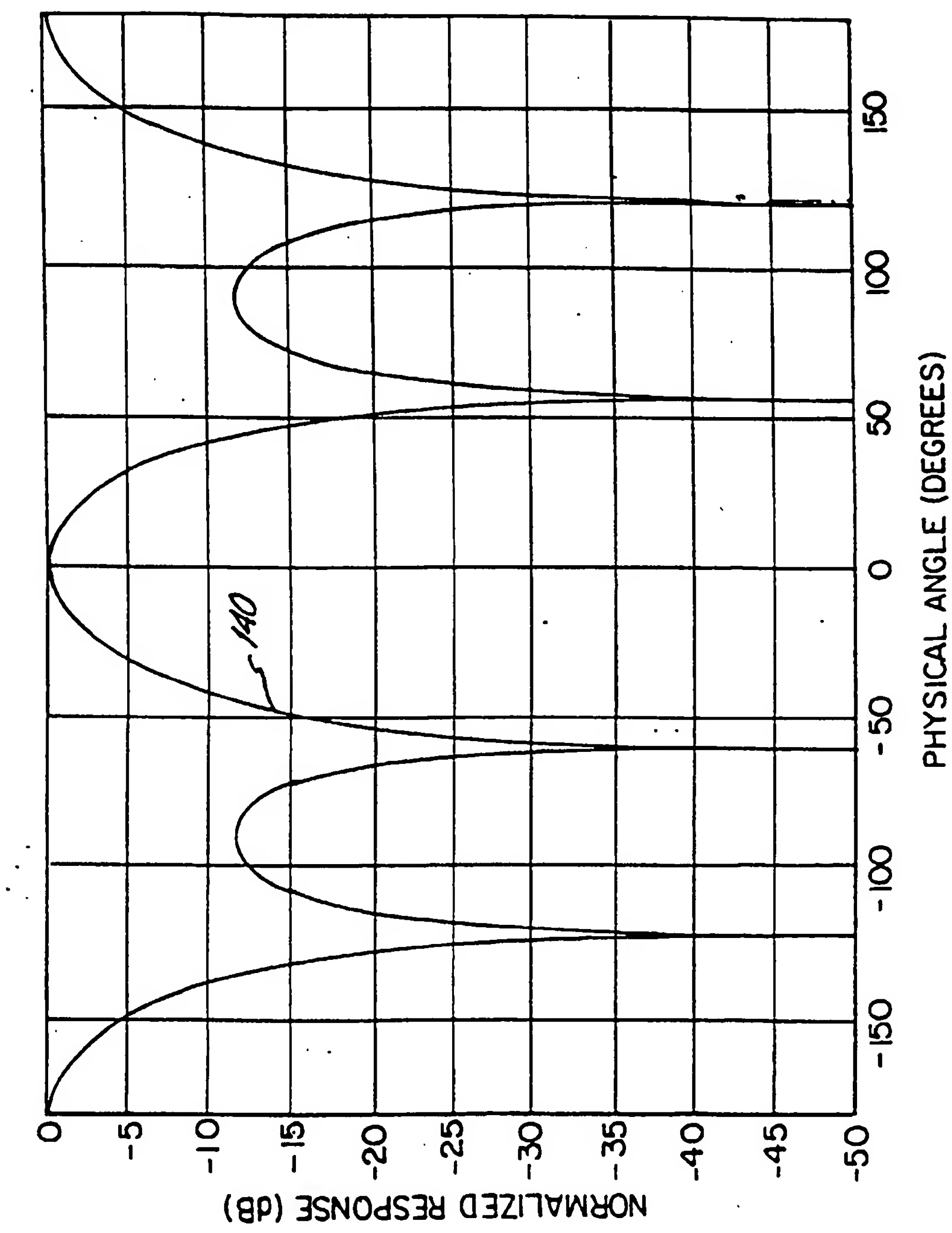

NORMALIZED RESPONSE (dB)
0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
PHYSICAL ANGLE (DEGREES)
-150
-100
-50
0
50
100
150
140

Fig. 12

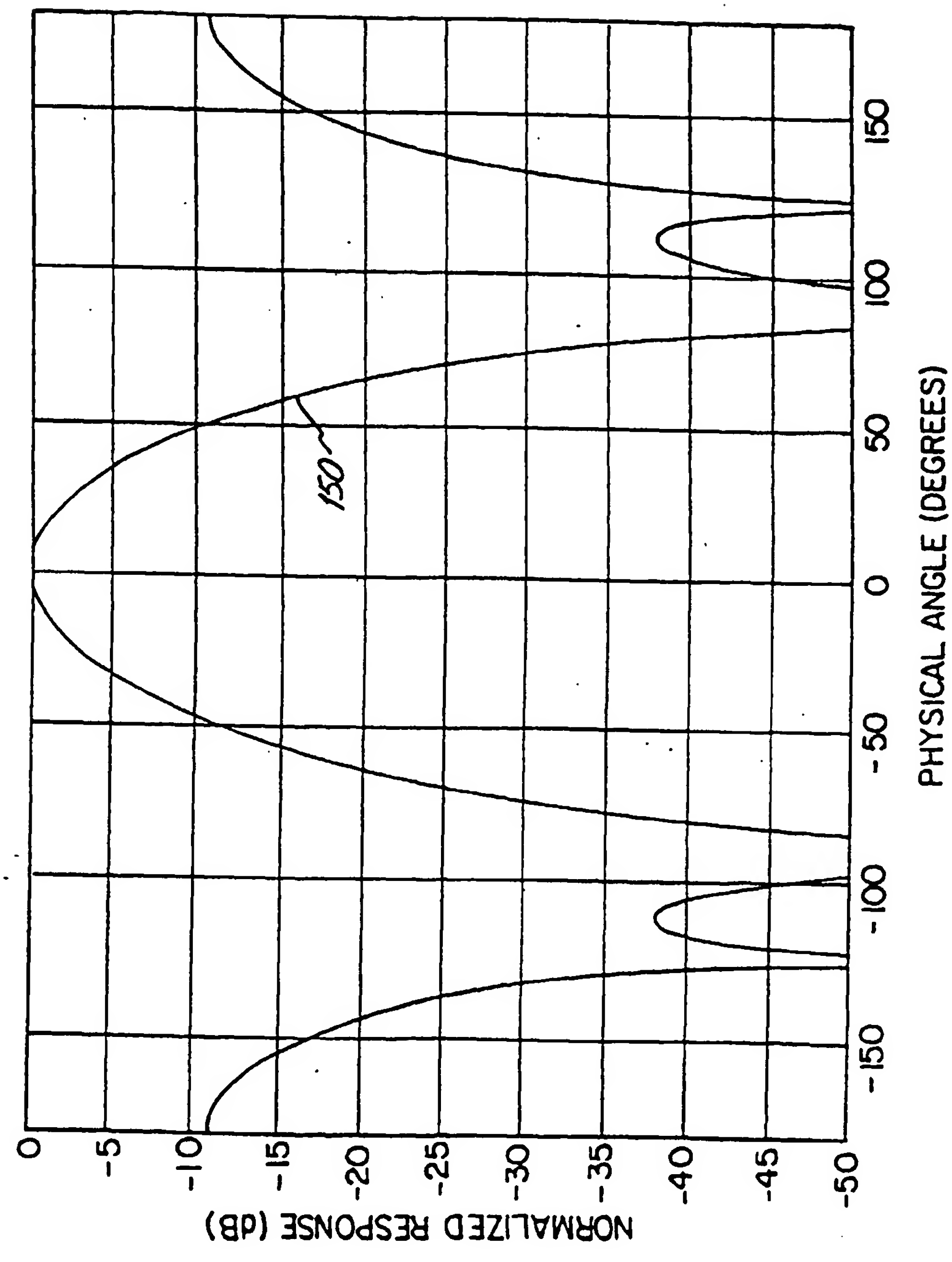

0
-5
-10
-15
-20
-25
-30
-35
-40
-45
-50
NORMALIZED RESPONSE (dB)
-150
-100
-50
0
50
100
150
PHYSICAL ANGLE (DEGREES)
150

Fig. 13

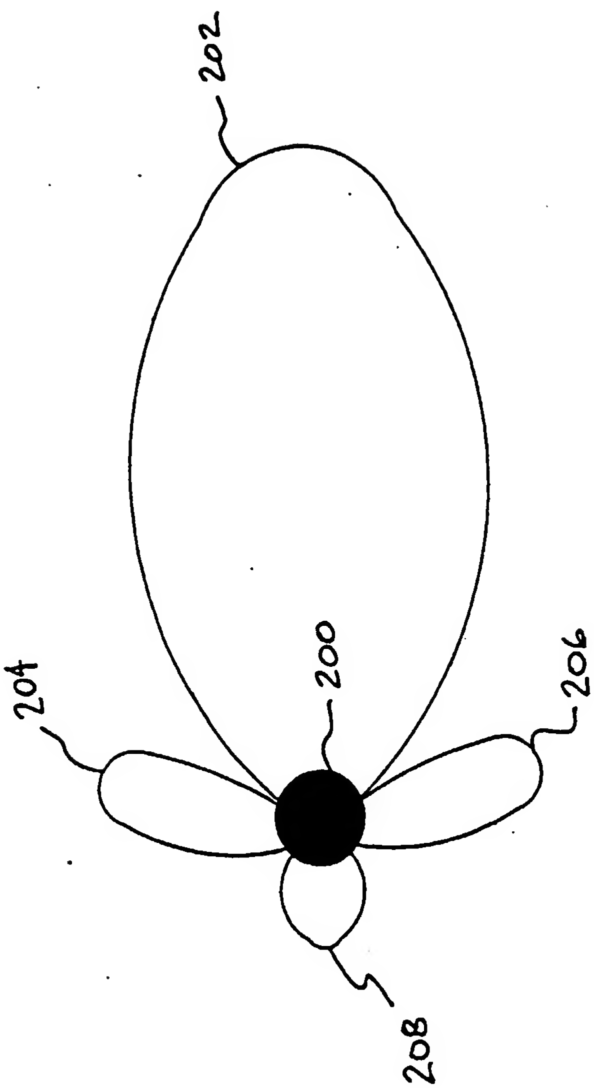



FIG. 14

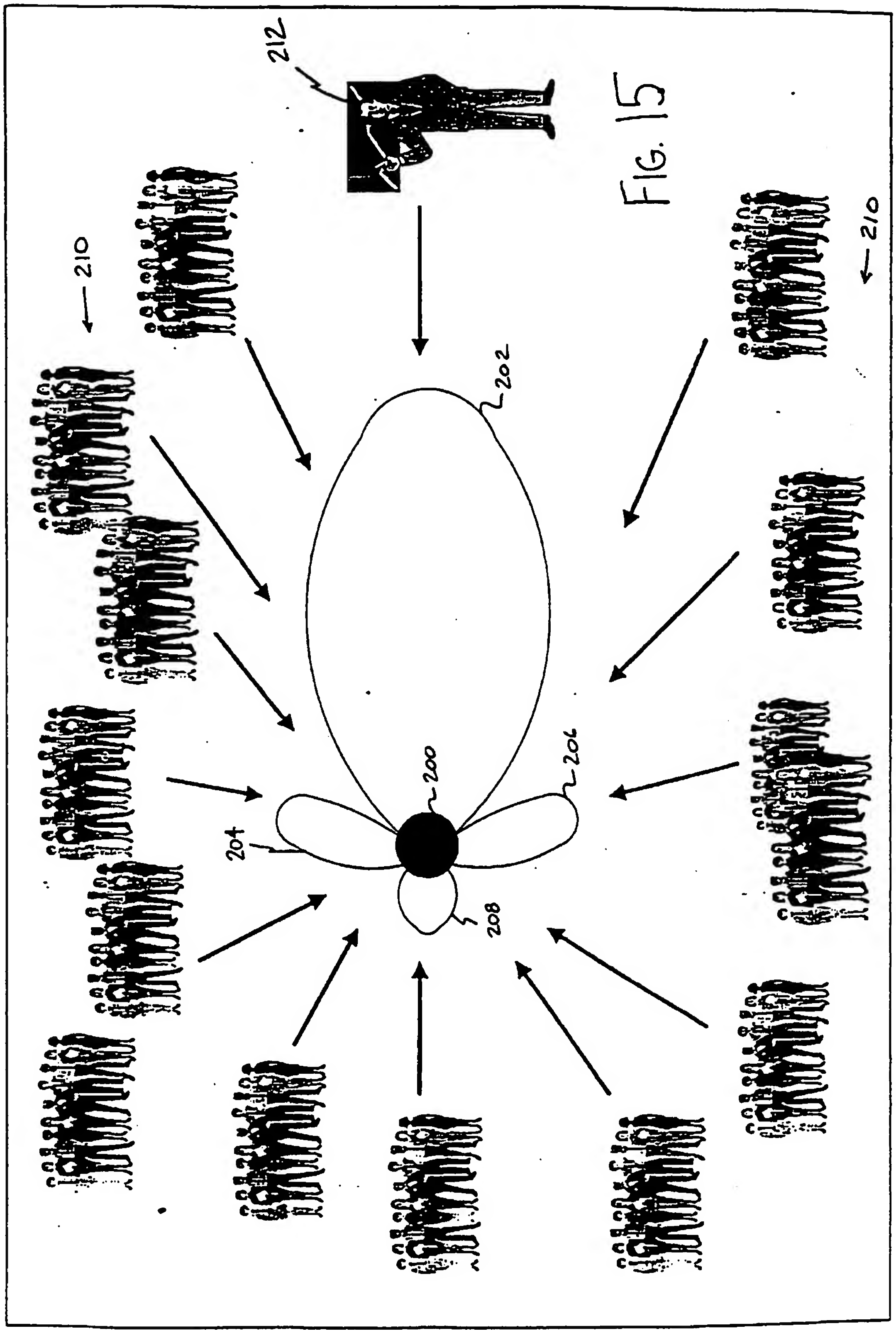
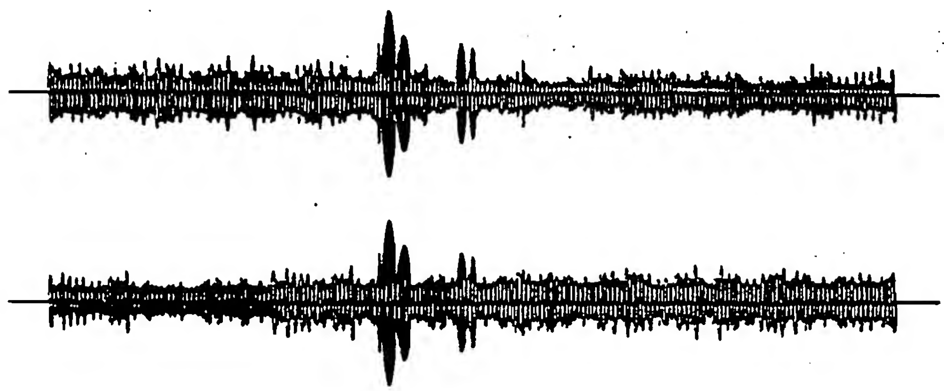



FIG. 15

Uncorrupted Signal

FIG. 16A



Left and Right Input Channels
(corrupted with random noise)

FIG. 16B

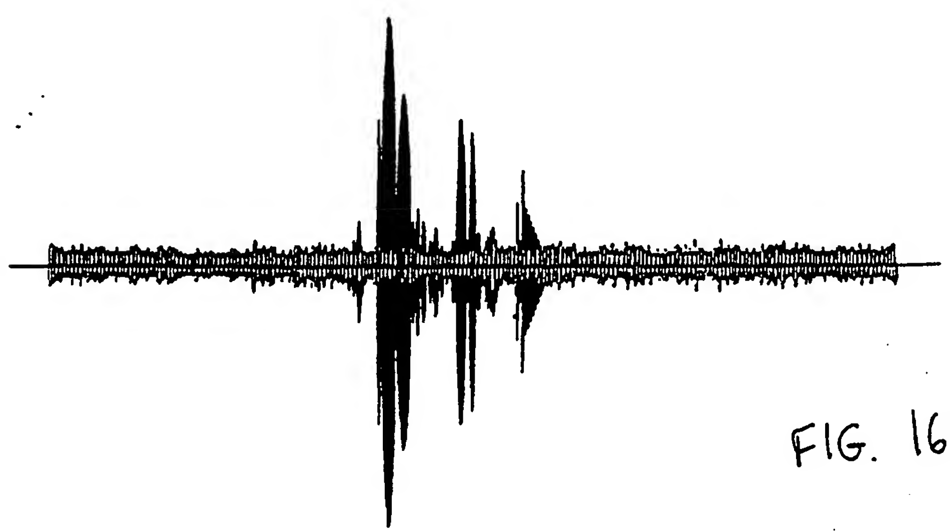


FIG. 16C

Beamformer Output

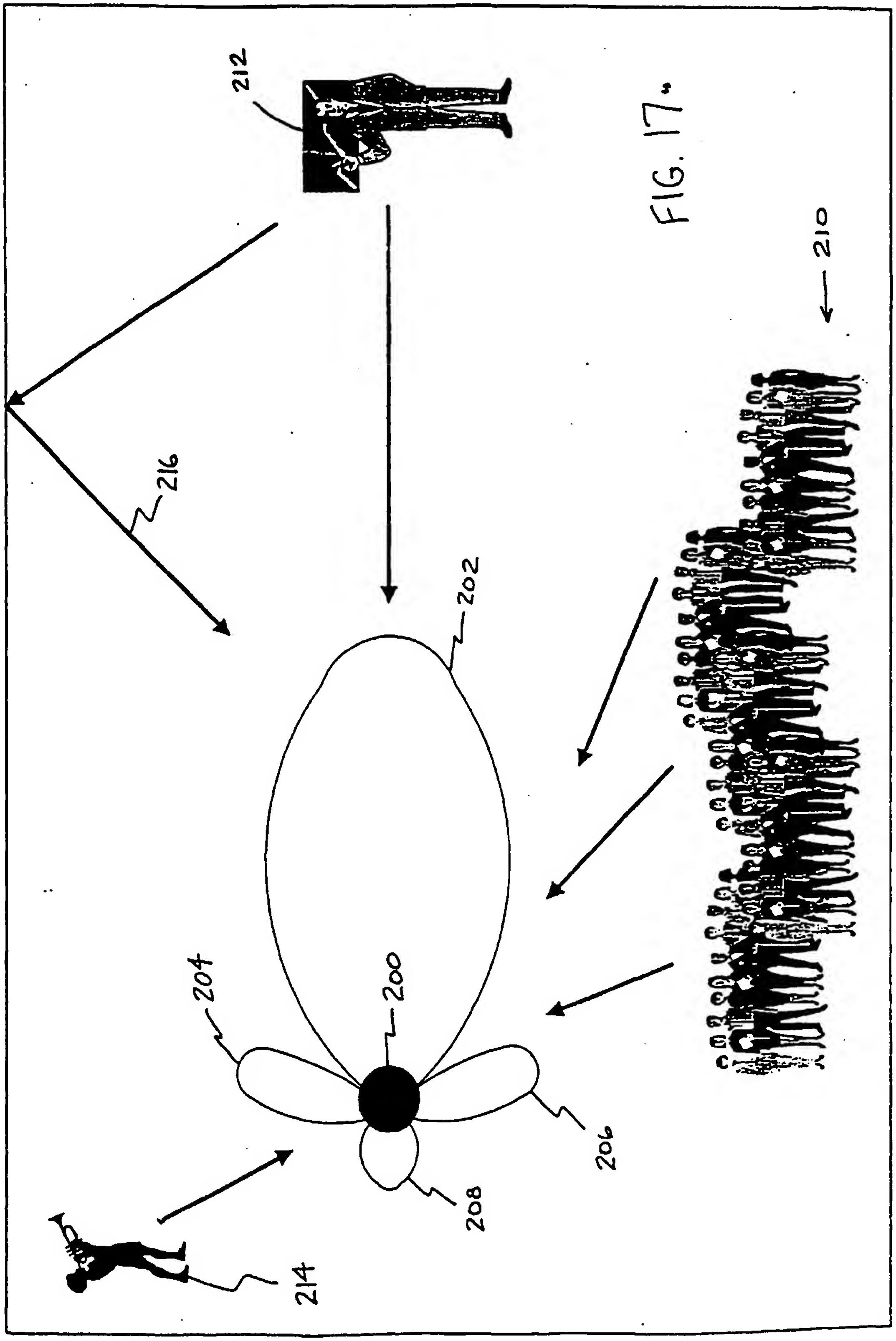



FIG. 17.

Uncorrupted Signal

FIG. 18A

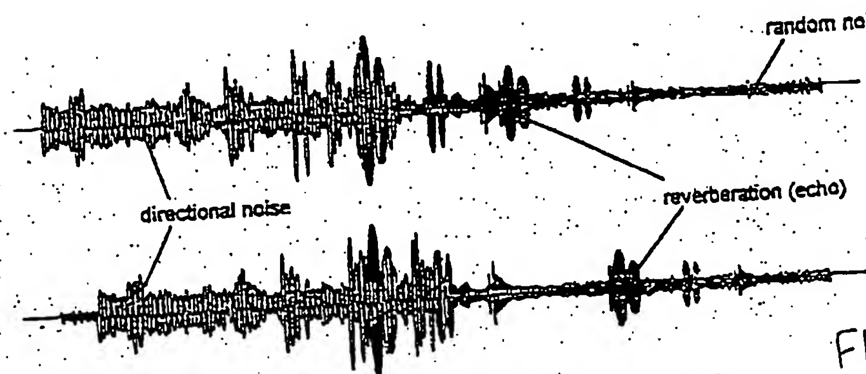



FIG. 18B

Left and Right Input Channels
(corrupted with random noise, directional noise and reverberation)

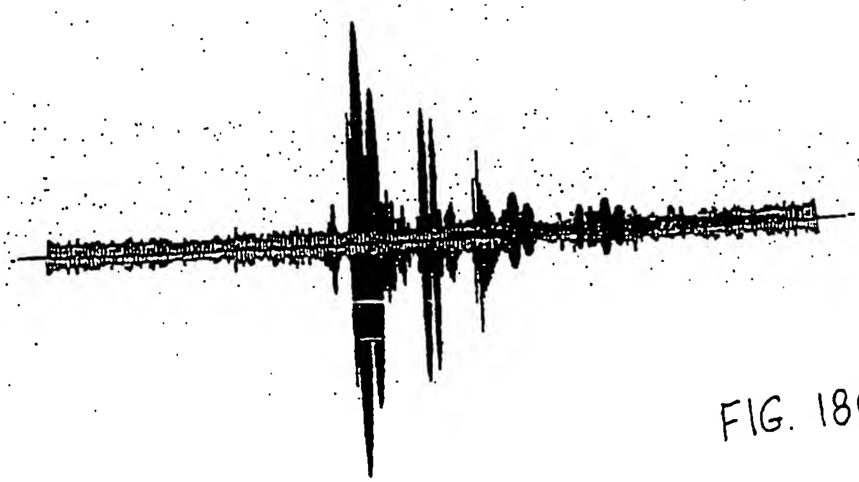


FIG. 18C

Beamformer Output